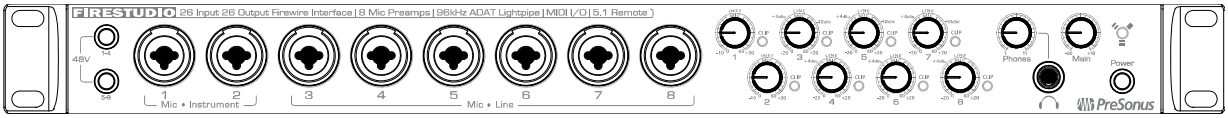
FIRESTUDIO

26 I/O 24-bit/96k Recording Interface with DSP Mixer/Router Control Console Application



User's Manual
Version 2.0

PreSonus

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TABLE OF CONTENTS

1 OVERVIEW

1.1 Introduction 3
1.2 Features 4
1.3 What is in the Box 5
1.4 System Requirements 6

2 OPERATION

2.1 Quick Start Up 7
2.1.1 Installation in Microsoft Windows 7
2.1.2 Installation in Mac OS X 9
2.1.3 Cubase LE – Device Setup 12
2.1.4 Cubase LE – Creating a Project 14
2.2 Sample Hookup Diagrams 16
2.2.1 Mobile Recording with the FireStudio and Two DigiMax FS 16
2.2.2 Full Band Set-up Using the MSR in Track Mode 17
2.3 Microphones 18
2.3.1 Condenser 18
2.3.2 Dynamic 18
2.3.3 USB and other types 18
2.4 A Brief Tutorial on Dynamics Processing 19
2.4.1 Common Questions Regarding Dynamics 19
2.4.2 Types of Dynamics Processing 20
2.4.3 The Vocabulary of Dynamics Processors 22
2.5 Digital Connections and Synchronization 24
2.5.1 What is ADAT optical? 24
2.5.2 What is S/MUX? 24
2.5.3 BNC Sync and Word Clock 24
2.5.4 Master/Slave and Multiple Digital Devices 25

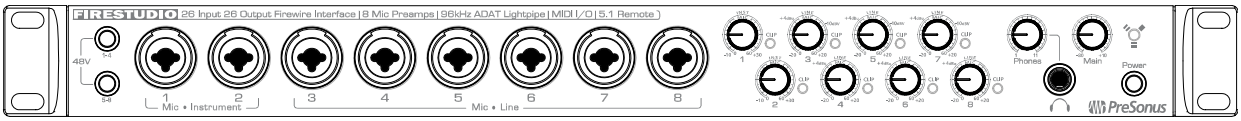
3 CONTROLS & CONNECTIONS

3.1 PreSonus FireStudio Control Console Application Introduction 26
3.1.1 FireStudio Control Console Application Icon 27
3.2 Hardware Settings Tab Introduction 28
3.2.1 FireStudio Settings Section 28
3.2.2 MSR Settings Section 30
3.2.3 Version Information Section 30
3.3 Output / Router Matrix Tab Introduction 31
3.3.1 Inputs and Playback Stream Routing 32
3.3.2 Mix Routing to Outputs 34
3.3.3 Output Faders 35
3.4 Mixer Tab Introduction 36
3.4.1 Channel Fader Controls 37
3.4.2 Master Fader 38
3.4.3 Quick Route Section 39
3.4.4 Naming Inputs and Mixes 40
3.5 Front Panel Layout 41
3.6 Back Panel Layout 43

4 TECHNICAL INFORMATION

4.1 Troubleshooting 47
4.2 Specifications 49

1.1 INTRODUCTION



Thank you for purchasing the PreSonus FireStudio. PreSonus Audio Electronics has designed the FireStudio utilizing high-grade components to ensure optimum performance that will last a lifetime. Loaded with 24-bit 96K converters, eight PreSonus X-MAX microphone preamplifiers, SPDIF, Dual SMUX 96k ADAT, BNC word clock, and MIDI I/O, as well as Cubase LE 48-track music production software and the optional Monitor Station Remote, the FireStudio is ready to go out of the box for professional quality computer recording. All you need is a computer with a firewire connection, a few microphones and cables along with your instruments and you are ready to record!

We encourage you to contact us at 225-216-7887 with any questions or comments you may have regarding your PreSonus FireStudio. PreSonus Audio Electronics is committed to constant product improvement, and we value your suggestions highly. We believe the best way to achieve our goal of constant product improvement is by listening to the real *experts*, our valued customers. We appreciate the support you have shown us through the purchase of this product.

We suggest you use this manual to familiarize yourself with the features, applications and correct connection procedure for your FireStudio before trying to connect it to your computer. This will hopefully alleviate any unforeseen issues that you may encounter during installation and set up.

Thank you, once again, for buying our product, and we hope you enjoy your FireStudio!

1.2 FEATURES

The FireStudio is a powerful and affordable computer recording system complete with everything needed for real-world recording applications. The FireStudio comes equipped with eight high-quality PreSonus X-MAX microphone preamps, MIDI I/O, SPDIF, Dual SMUX 96k ADAT, BNC word clock and rock solid drivers, as well as a plethora of music recording and production software.

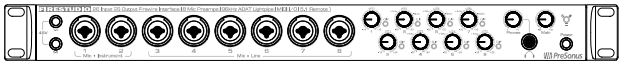
The FireStudio includes PreSonus ProPak Software Suite with Cubase LE and over 2 GB of plug-ins, drum loops and samples – giving you everything you need for professional music recording and production.

Summary of features

- High-speed FireWire (IEEE 1394)
- 24-bit / 96 kHz sampling rate
- 26 simultaneous inputs and outputs
- 8 Class A XMAX microphone preamplifiers
- 2 Insert channels
- 8 Analog line inputs
- 2 Instrument inputs
- 16 Channels of ADAT optical I/O (8 channels via 96k dual S/MUX)
- 10 balanced TRS outputs
- S/PDIF I/O
- MIDI I/O
- 36x18 FireStudio Control Console DSP mixer
- JetPLL jitter control for improved imaging and clarity
- High performance A/D D/A converters
- Cubase LE Music Production Software included
- PreSonus ProPak Software Suite (over 2 GB of plug-ins, loops and samples)
- OS X and Windows compatible
- MSR Monitor Station Remote (Optional)

1.3 WHAT IS IN THE BOX

Your FireStudio package contains the following:



- FireStudio Recording Interface

- 6’ 6-pin to 6-pin FireWire cable

- Power Supply

- Software installation discs:
  - PreSonus FireStudio Installation Drivers
  - Cubase LE
  - ProPak Software

- PreSonus Warranty Card

1.4 SYSTEM REQUIREMENTS

Below are the minimum computer system requirements for your FireStudio.

Macintosh

- OS: Mac OS X 10.4.x or later
- Computer: Apple Macintosh series with FireWire 400 port
- CPU/Clock: PowerPC G4/1 GHz or higher (Intel/Dual 1 GHz recommended)
- Memory (RAM): 512 MB (1 GB or more recommended)

Windows

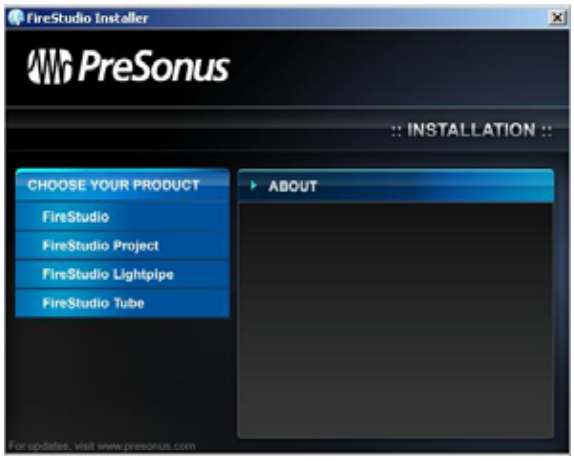
- OS: 32-bit Microsoft Windows Vista or XP (with at least SP1)
- Computer: Windows compatible computer with FireWire 400 port.
- CPU/Clock: Pentium or AMD with 1 GHz or higher (Dual 1 GHz recommended)
- Memory (RAM): 512 MB (1 GB or more recommended)

NOTE: The speed of your processor, amount of RAM and size & speed of your hard drives will greatly affect the overall performance of your recording system. A more powerful system (faster processor with more RAM) allows for lower latency (signal delay) and better overall performance.

2.1 QUICK START UP

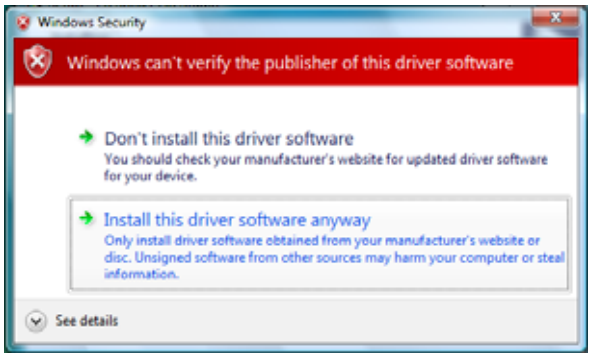
2.1.1 Installation in Microsoft Windows

After inserting the Installation CD into your CD-ROM drive, the FireStudio installer will take you through each step of the installation process. Please read each message carefully – ensuring especially that you do not connect your FireStudio early.



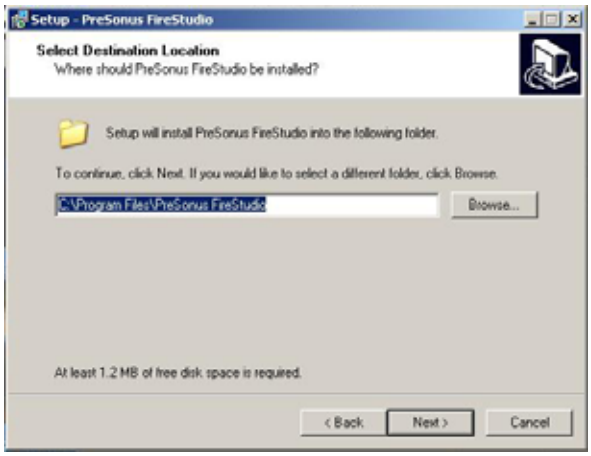
1) The FireStudio Installer was written to be as simple and easy-to-follow as possible. Please read each message carefully to ensure the FireStudio is properly installed.

   Before continuing the FireStudio Installation Setup, please close all multimedia applications and disconnect your FireStudio from your computer.

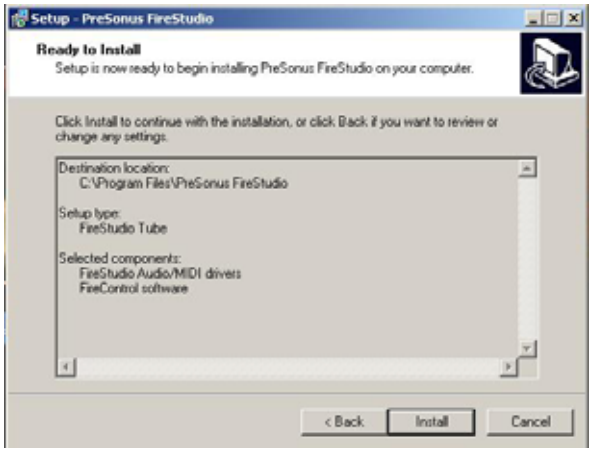

Windows Security
Windows can't verify the publisher of this driver software
Don't install this driver software
You should check your manufacturer's website for updated driver software for your device.
Install this driver software anyway
Only install driver software obtained from your manufacturer's website or disc. Unsigned software from other sources may harm your computer or steal information.
See details

2) If you see any Windows Security alerts, click "Install this driver software anyway" (Vista) or "Continue anyway" (XP).

3) This installer can be used for the entire FireStudio family of interface products. Select "FireStudio" from the menu on the left of the installer window


Setup - PreSonus FireStudio
Select Destination Location
Where should PreSonus FireStudio be installed?
Setup will install PreSonus FireStudio into the following folder.
To continue, click Next. If you would like to select a different folder, click Browse.
C:\Program Files\PreSonus FireStudio
Browse...
At least 1.2 MB of free disk space is required.
< Back
Next >
Cancel

4) The installer will now display the directory path into which the FireStudio Control Console application will be installed. Click "Next"


Setup - PreSonus FireStudio
Ready to Install
Setup is now ready to begin installing PreSonus FireStudio on your computer.
Click Install to continue with the installation, or click Back if you want to review or change any settings.
Destination location:
C:\Program Files\PreSonus FireStudio
Setup type:
FireStudio Tube
Selected components:
FireStudio Audio/MIDI drivers
FireControl software
< Back
Install
Cancel

5) The installer is now ready to begin. Click "Install" to copy the correct components to your computer.

Setup - PreSonus FireStudio
PreSonus
:: INSTALLATION ::
Completing the PreSonus FireStudio Setup Wizard
To complete the installation of PreSonus FireStudio, Setup must restart your computer. Would you like to restart now?
Yes, restart the computer now
No, I will restart the computer later
Finish

6) The installer will now prompt you to reboot your computer to complete the installation. Click "Finish" to automatically restart your PC. Once your computer has rebooted, connect the FireStudio. When the Found New Hardware wizard launches, follow the recommended steps. When the sync light remains solid blue, your FireStudio is synced to your computer and ready to use!

Please Note: When launching the FireStudio Control Console, you may be directed to update your FireStudio's firmware. Follow the onscreen instructions. Do not disconnect or power down your FireStudio during the firmware update.

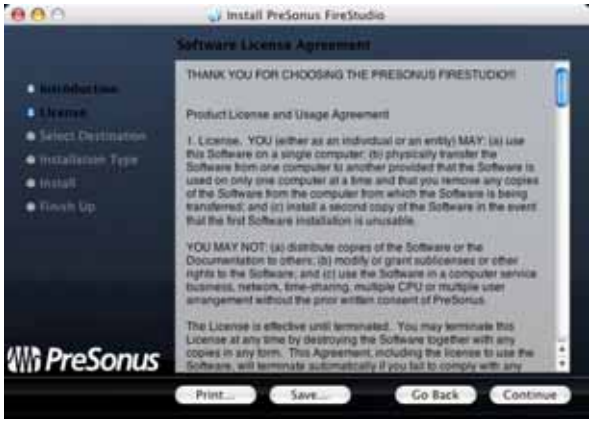
2.1.2 Installation in Mac OS X

After inserting the Installation CD into your disc drive, browse the disc and run the FireStudio installer, which will take you through each step of the installation process. Please read each message carefully – ensuring especially that you do not connect your FireStudio early.

1) To begin installing the FireStudio driver on your Mac, double click on the FireStudio logo.

2) You will be directed to the welcome screen.

   Click "Continue"



3) Take a moment to review the license agreement and click "Continue".

Install PreSonus FireStudio
To continue installing the software, you must agree to the terms of the software license agreement.
Click Agree to continue or click Disagree to cancel the installation.
Disagree
Agree
PRESONUS IS A REGISTRED TRADEMARK IN USA.
ALL FEATURES AND SPECIFICATIONS SUBJECT TO CHANGE WITHOUT NOTICE.
PreSonus
Print...
Save...
Go Back
Continue

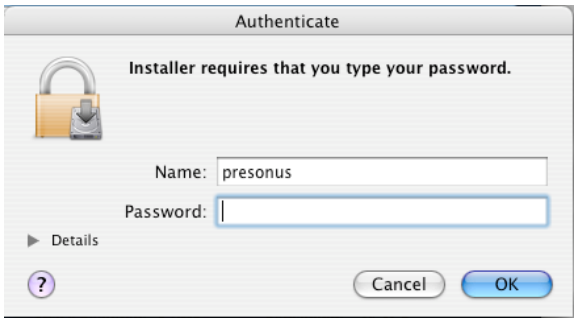
4) Click "Agree" to continue installation.

Install PreSonus FireStudio
Select a Destination
Select a destination volume to install the PreSonus FireStudio software.
Macintosh HD
74.2GB (24.3GB Free)
You have chosen to install this software on the volume "Macintosh HD."
PreSonus
Go Back
Continue

5) You will be directed to choose the hard drive onto which you want to install the drivers and FireStudio Control Console Software. You must choose your system drive. Storage drives and partitions cannot be used as hosts for the driver.

6) The FireStudio installer contains the installations for the entire FireStudio family of interfaces. Select "FireStudio" from the list. A " – " will appear in the box to the left.

Click "Upgrade"



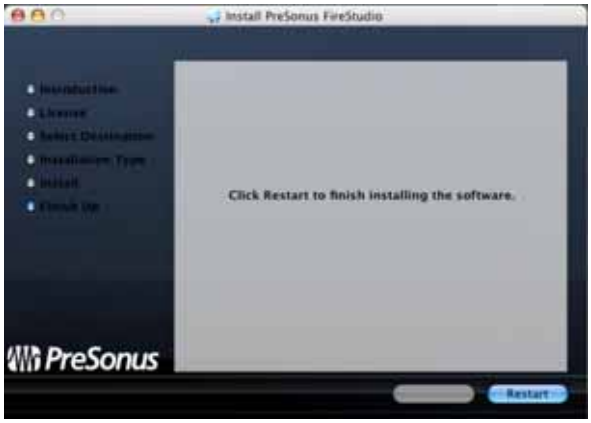
7) The FireStudio installer requires that your user password be entered as a security measure.

Enter your password and click "OK"

Install PreSonus FireStudio
Installing this software requires you to restart your computer when the installation is done. Are you sure you want to install the software now?
Cancel
Continue Installation
Drivers and Applications for the FireStudio Project and FireStudio Tube 24-bit/96k professional recording systems.
PreSonus
Easy Install
Go Back
Upgrade

8) You will be notified that the FireStudio installation requires that your Mac be restarted to finish the installation process.

Click "Continue Installation"



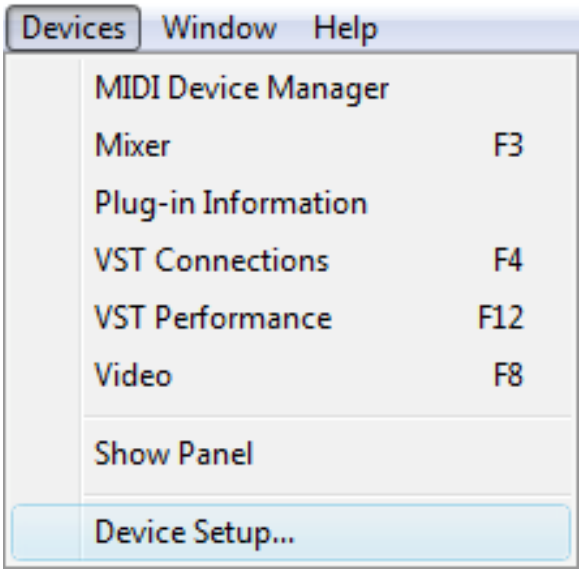
9) Click "Restart" to automatically reboot your Mac.

After rebooting your computer, connect your FireStudio to your computer with the included firewire cable. When the sync light remains solid blue, your FireStudio is synced to your computer and ready to use!

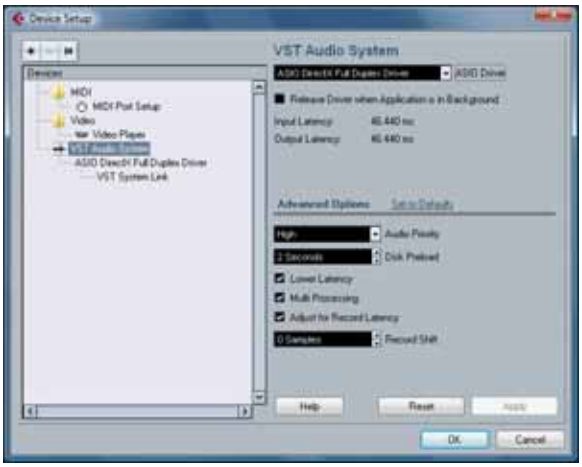
2.1.3 Cubase LE – Device Setup

Once you have installed the FireStudio drivers and connected your FireStudio, you can use the Cubase LE Music Creation and Production System software included with your FireStudio to begin recording, mixing and producing your music. Insert the Cubase LE installation CD into your computer and run the installer.

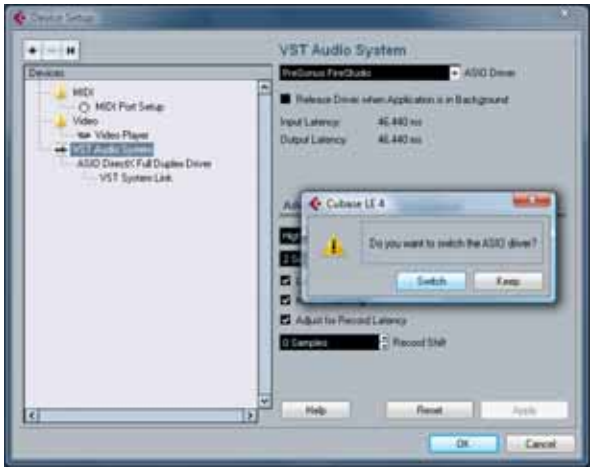
Then, launch Cubase LE and follow these steps to begin recording with your FireStudio:



1) *S*elect 'Device Setup…' from the *Devices* menu.

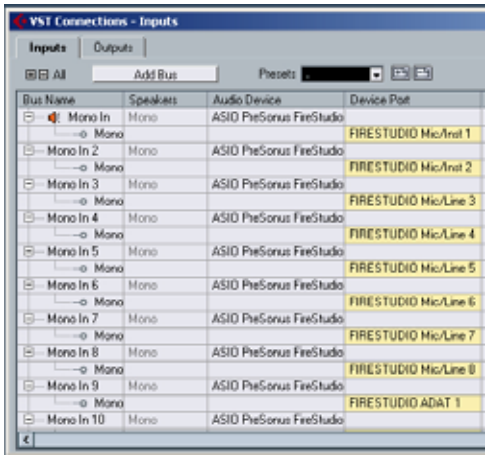


2) Select 'VST Audio System' from the *Devices* column in Device Setup.



3) Select "PreSonus FireStudio" from the *ASIO* Driver dropdown list. A pop-up window will appear asking if you'd like to switch the ASIO driver. Click 'Switch' to begin using the FireStudio Driver.

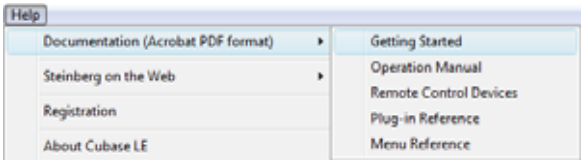
4) Go to Devices>VST Connections to set up your FireStudio's input busses. Click on Add Bus. You can choose between mono and stereo busses. Select the type and number of busses you'd like to create. In this example, we've created 26 mono inputs to access all of the inputs on the FireStudio.



5) You will now see all of your input busses on the Inputs tab. Cubase automatically assigns the FireStudio inputs to the respective bus (i.e. FireStudio Input 1 to Mono In 1, etc.).

Click on the Ouputs tab and follow the same steps to create your Output busses.

Please Note: you can change the bus routing to another FireStudio input or output at any time by simply left clicking on the Device Port name. You can also customize your bus names by double clicking on them.

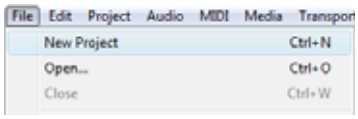


For more help on using Cubase LE, please read the *Getting Started* and *Operation Manual* help documents located in Help > Documentation.

2.1.4 Cubase LE – Creating a Project

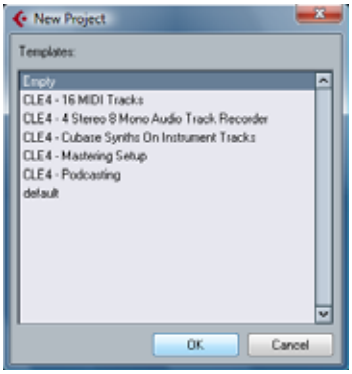
To begin recording in Cubase LE, you first need to create a project. Once you have the project environment, you can begin adding audio and MIDI tracks to record and edit on. Follow these simple steps to begin recording your first audio in Cubase LE:

1) Plug an XLR microphone or instrument into FireStudio channel one and turn on 48V phantom power if needed for your microphone (typically only for condenser microphones).



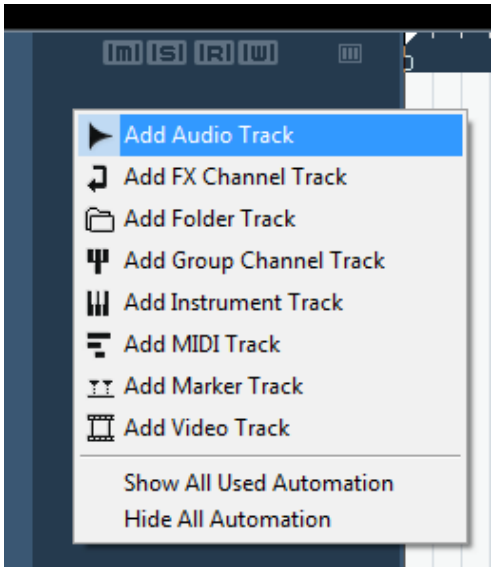
2) Enter the *File* menu and select 'New Project'.

You can also use the keyboard shortcut Ctrl+N.



3) Select 'Empty' when prompted for a Template.

Click 'OK'.



4) Right-click in the track space within the new project and select 'Add Audio Track' from the dropdown menu which appears.

You can also add a track by entering the Project *menu and selecting 'Add Track'.*

5) When the Add Audio Track dialogue appears, choose the number of audio tracks you wish to add, whether they will be Mono or Stereo and then click 'OK'.

If you have two audio signals making up a stereo pair (i.e. keyboard, FX processor, cd player, etc.), you should use a Stereo Configuration.

Otherwise, if you have a single, mono audio source (i.e. microphone or instrument), you will want to choose a Mono Configuration.

6) Click the Record Enable button to arm the new track for recording.

7) Click the Monitor button to hear the input through the FireStudio's 1/2 and Main CR outputs.

8) Select FireStudio input 'Stereo In > Left'.

   When choosing a mono from a stereo pair, the "left" channel is typically the odd-numbered channel, and the "right" channel is typically the even-numbered channel (i.e. Left is 1; Right is 2).

   Press <F4> on your keyboard to enter the VST Connections menu for more details on your specific input and output routing.

9) Turn up the gain control knob on the front panel of the FireStudio for channel 1 while speaking/singing into the microphone or playing your instrument. You should see the input meter in Cubase LE react to the input. Adjust the gain so the input level is near its maximum **without clipping**.

10) Connect a set of headphones to the FireStudio headphone output. You may also wish to connect monitors to the FireStudio's Main Outs.

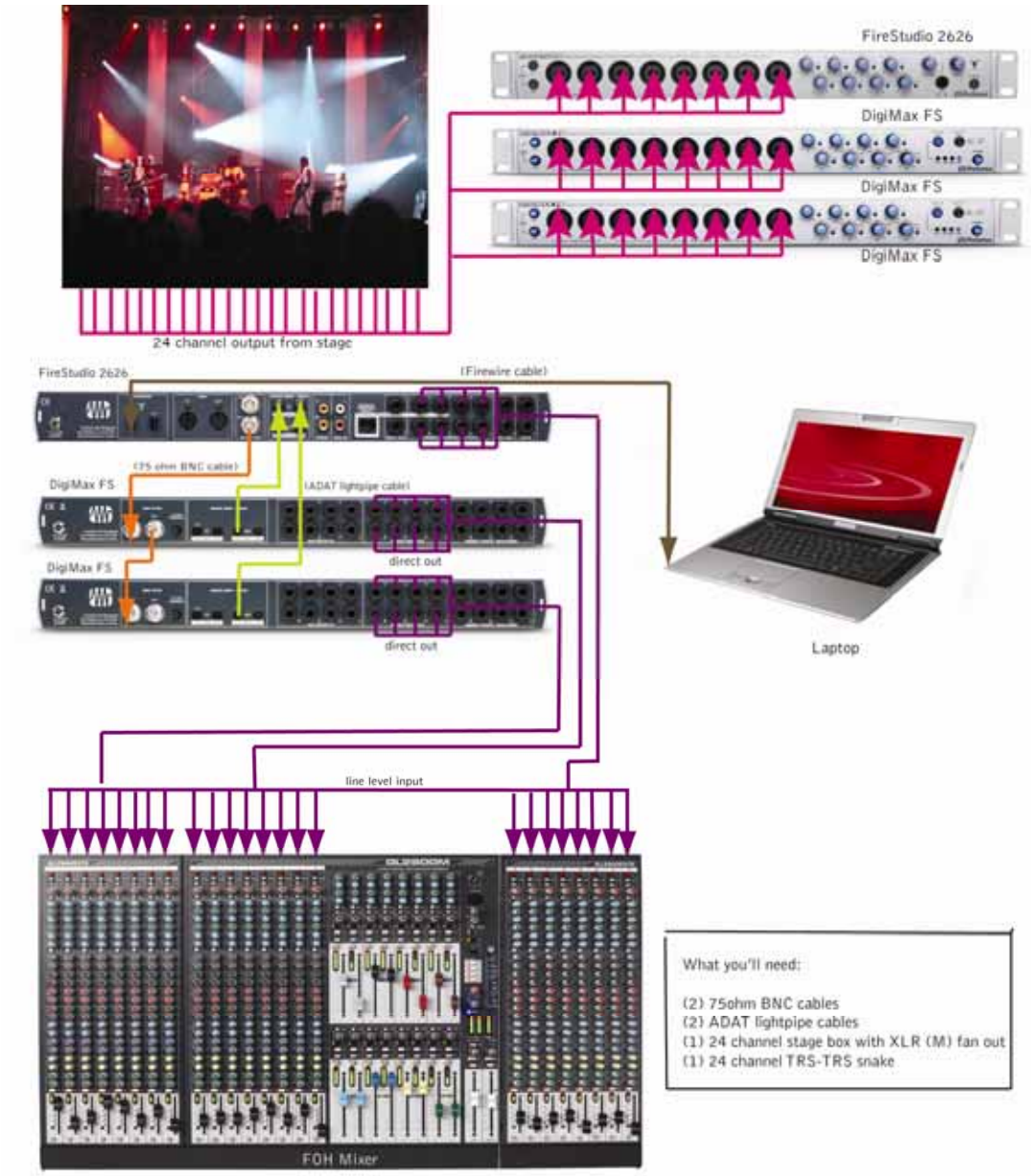
11) You are now ready to record.

Cubase LE help is available by pressing <F1> on your keyboard or online by visiting www.steinberg.net.

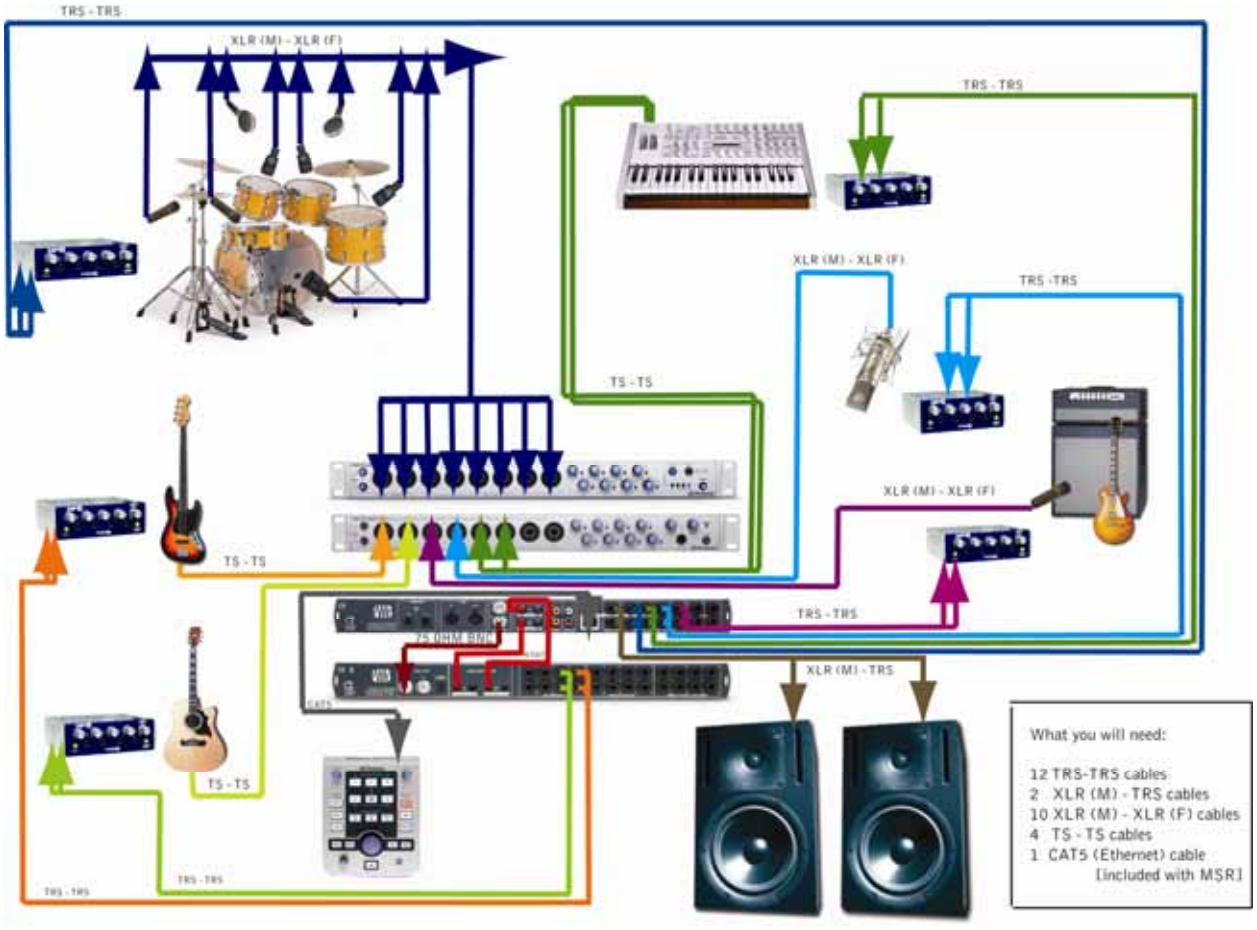
2.2 SAMPLE HOOK UP DIAGRAMS

With the FireStudio, you can simultaneously record and play back up to 26 channels. Since it is loaded with eight preamplifiers, you can plug in eight microphones to the FireStudio. Using the ADAT optical and S/PDIF inputs, you can connect external A/D devices like the DigiMax FS. This makes recording extremely easy. All you need are a few microphones, some cables to connect them, a musician (or two or three or more) and the creative energy to bring it all together.

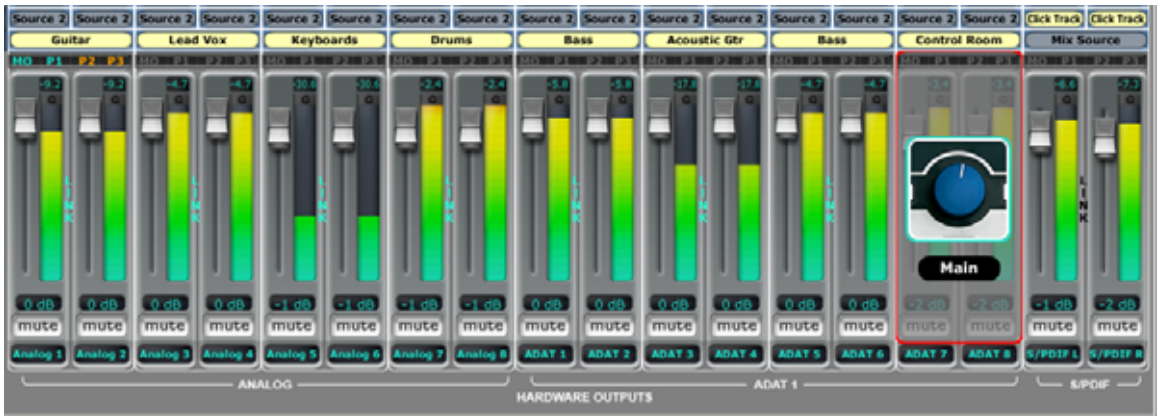
2.2.1 Mobile Recording with the FireStudio and Two DigiMax FS



2.2.2 Full Band Set-Up Using the MSR in Track Mode



Please note: This set-up can be used with or without an MSR connected. This example implies that seven monitor mixes have been created in the FireStudio Control Console and routed to the 8 General Purpose outputs of the FireStudio and 4 of the DAC outputs on the DigiMax FS in stereo pairs. The seventh mix is assigned to an unused output and copied to the Main Outputs for control room monitoring (see below). By connecting an ADAT cable to both the input and the output of the DigiMax FS, it can be used to expand the outputs of the FireStudio as well as creating additional analog inputs.



2.3 MICROPHONES

The FireStudio works with many microphones including dynamic, ribbon and condenser microphones.

2.3.1 Condenser

Condenser microphones tend to generate a high-quality audio signal and are one of the most popular microphone choices for today's studio recording applications. Condenser microphones require a power source, which can be provided from a small battery, external power supply or from microphone inputs as *phantom power.* The FireStudio sends phantom power from the XLR inputs only.

2.3.2 Dynamic

Dynamic microphones are possibly the most widely used microphone type – especially in live shows. They are relatively inexpensive, resistant to physical damage and typically handle high sound pressure levels (SPL) very well. Unlike condenser microphones, dynamic microphones do not require a power source and, in most cases, phantom power has no effect on a dynamic microphone's audio quality or sensitivity.

Dynamic microphones, especially ribbon microphones, tend to generate low output voltages, so they typically need more preamp gain than a condenser microphone.

Ribbon

Ribbon microphones are a special type of dynamic microphone and get their name from the thin metal ribbon used in their design. Ribbon microphones have very high quality sound reproduction qualities – especially with higher frequencies sounds. However, they are very fragile and typically cannot handle high SPL's.

The most important thing to note about Ribbon microphones is that nearly all Ribbon Microphones ***do not require phantom power.*** In fact, unless a Ribbon microphone specifically calls for phantom power, sending phantom power to a ribbon microphone will destroy it – usually beyond repair.

2.3.3 USB and other types

There are a vast number of microphone types available, and as technology increases, it is very likely more will surface. One type of microphone to emerge recently is the USB microphone. Most USB microphones have their own built-in preamp and use drivers separate from the FireStudio drivers. Because USB microphones are, in effect, an audio interface, it is not recommended to use them with the FireStudio as the likelihood of conflicting drivers is great.

If you are using a new or non-standard type of microphone (i.e., USB, headset, laser, MEMS, etc.), please consult your microphone's user's manual for power requirement or compatibility information.

Regardless of the microphone type you are using, we recommend reading your microphone's user's manual thoroughly before engaging phantom power or if any other usage questions may arise.

2.4 A BRIEF TUTORIAL ON DYANAMICS PROCESSING

The first two channels on the front panel of the FireStudio feature insert points for use with external dynamics processors. This section is an excerpt from brief tutorial on dynamics processing written by PreSonus President and Chief Technical Officer, Jim Odom. It is included to help you get the most out of your FireStudio. This tutorial will take you through the basics of dynamics processing as well as explain the various types of dynamics processors.

2.4.1 Common Questions Regarding Dynamics

What is dynamic range?

Dynamic range can be defined as the distance between the loudest possible level to the lowest possible level. For example, if a processor states that the maximum input level before distortion is +24dBu and the output noise floor is -92dBu, then the processor has a total dynamic range of 24 + 92 = 116dB. The average dynamic range of an orchestral performance can range from -50dBu to +10dBu on average. This equates to a 60dB dynamic range. 60dB may not appear to be a large dynamic range but do the math and you'll discover that +10dBu is 1000 times louder than -50dBu! Rock music on the other hand has a much smaller dynamic range, typically -10dBu to +10dBu, or 20dB. This makes mixing the various signals of a rock performance together a much more tedious task.

Why do we need compression?

Consider the previous discussion: You are mixing a rock performance with an average dynamic range of 20dB. You wish to add an un-compressed vocal to the mix. The average dynamic range of an uncompressed vocal is around 40dB. In other words a vocal performance can go from -30dBu to +10dBu. The passages that are +10dBu and higher will be heard over the mix. However, the passages that are at -30dBu and below will never be heard over the roar of the rest of the mix. A compressor can be used in this situation to reduce (compress) the dynamic range of the vocal to around 10dB. The vocal can now be placed at around +5dBu. At this level, the dynamic range of the vocal is from 0dBu to +10dBu. The lower level phrases will now be well above the lower level of the mix and louder phrases will not overpower the mix, allowing the vocal to 'sit in the track'.

The same discussion can be made about any instrument in the mix. Each instrument has its place and a good compressor can assist the engineer in the overall blend of each instrument.

Does every instrument need compression?

This question may lead many folks to say 'absolutely not, overcompression is horrible'. That statement can be qualified by defining 'overcompression'. The term itself, 'overcompression' must have been derived from the fact that you can hear the compressor working. A well designed and properly adjusted compressor should not be audible! Therefore, the overcompressed sound is likely to be an improper adjustment on a particular instrument.

Why do the best consoles in the world put compressors on every channel? The answer is simply that most instruments need some form of compression, oftentimes very subtle, to be properly heard in a mix.

Why do you need noise gates?

Consider the compressed vocal example above and you now have a 20dB dynamic range for the vocal channel. Problems arise when there is noise or instruments in the background of the vocal mic that became more audible after the lower end of the dynamic range was raised (air conditioner, loud drummer, etc). You might attempt to mute the vocal between phrases in an attempt to remove the unwanted signals; however this would probably end disastrously. A better method is to use a noise gate. The noise gate threshold could be set at the bottom of the dynamic range of the vocal, say -10dBu, such that the gate would 'close' out the unwanted signals between the phrases.

If you have ever mixed live you know well the problem cymbals can add to your job by bleeding through your tom mics. As soon as you add some highs to get some snap out of the tom the cymbals come crashing through, placing the horn drivers into a small orbit. Gating those toms so that the cymbals no longer ring through the tom mics will give you an enormous boost in cleaning up the overall mix.

2.4.2 Types of Dynamics Processing

Dynamics processing is the process of altering the dynamic range or levels of a signal thereby enhancing the ability of a live sound system or recording device to handle the signal without distortion or noise, and aiding in placing the signal in the overall mix.

Compression / Limiting

Punch, apparent loudness, presence…just three of many terms used to describe the effects of compression/limiting. Compression and limiting are forms of dynamic range (volume) control. Audio signals have very wide peak to average signal level ratios (sometimes referred to as dynamic range which is the difference between the loudest level and the softest level). The peak signal can cause overload in the audio recording or reproduction chain resulting in signal distortion.

A compressor/limiter is a type of amplifier in which gain is dependent on the signal level passing through it. You can set the maximum level a compressor/limiter allows to pass through, thereby causing automatic gain reduction above some predetermined signal level or threshold. Compression refers, basically, to the ability to reduce the output level of an audio signal by a fixed ratio relative to the input. It is useful for lowering the dynamic range of an instrument or vocal, making it easier to record without distorting the recorder. It also assists in the mixing process by reducing the amount of level changes needed for a particular instrument.

Take, for example, a vocalist who moves around in front of the microphone while performing, making the output level vary up and down unnaturally. A compressor can be applied to the signal to help correct this recording problem by reducing the 'louder' passages enough to be compatible with the overall performance.
How severely the compressor reduces the signal is determined by the compression ratio and compression threshold. A ratio of 2:1 or less is considered mild compression, reducing the output by a factor of two for signals greater than the compression threshold. Ratios above 10:1 are considered hard limiting.

Limiting refers to the point at which the signal is restrained from going any louder at the output. The level of input signal at which the output is reduced is determined by the compression threshold. As the compression threshold is lowered, more and more of the input signal is compressed (assuming a nominal input signal level). Care must be taken not to 'over compress' a signal. Too much compression destroys the acoustic dynamic response of a performance. ('Over compression', however, is used by some engineers as an effect, and with killer results!)

Compressor / Limiters are commonly used for many audio applications. For example:

- A kick drum can get lost in a wall of electric guitars. No matter how much level is increased, the kick drum stays 'lost in the mud'. Add a touch of compression and tighten up that kick drum sound allowing it to 'punch' through without having to crank the level way up.

- A vocal performance usually has a wide dynamic range. Transients (the very loudest portion of the signal) can be far outside the average level of the vocal signal. It is extremely difficult to ride the level with a console fader. A compressor/limiter automatically controls gain without altering the subtleties of the performance.

- A solo guitar can seem to be masked by the rhythm guitars. Compression can make your 'lead' soar above the track without shoving the fader through the roof.

- Bass guitar can be difficult to record. A consistent level with good attack can be achieved with proper compression. Your bass doesn't have to be washed out in the low end of the mix. Let the compressor/limiter give your bass the punch it needs to drive the bottom of the mix.

Expansion

There are two basic types of expansion: dynamic and downward. Expansion increases the dynamic range or level of a signal after the signal crosses the expansion threshold. Dynamic expansion is basically the opposite of compression. In fact, broadcasters use dynamic expansion to 'undue' compression before transmitting the audio signal. This is commonly referred to as 'companding' or COMPression followed by expANDING.

By far the most common use of expansion is downward expansion. In contrast to compression, which decreases the level of a signal after rising above the compression threshold, expansion decreases the level of a signal after the signal goes below the expansion threshold. The amount of level reduction is determined by the expansion ratio. For example, a 2:1 expansion ratio reduces the level of a signal by a factor of two. (e.g. if a level drops 5dB below the expansion threshold, the expander will reduce it to 10dB below the threshold.)

Commonly used as noise reduction, expansion is very effective as a simple noise gate. The major difference between expansion and noise gating is the fact that expansion is dependent on the signal level after crossing the threshold, whereas a noise gate works independent of a signal's level after crossing the threshold.

Noise Gating

Noise gating is the process of removing unwanted sounds from a signal by attenuating all signals below a set threshold. As described above, the 'gate' works independent of the audio signal after being 'triggered' by the signal crossing the gate threshold. The gate will remain open as long as the signal is above the threshold. How fast the gate opens to let the 'good' signal through is determined by the attack time. How long the gate stays open after the signal has gone below the threshold is determined by the hold time. How fast the gate closes is determined by the release. How much the gate attenuates the unwanted signal while closed is determined by the range.

2.4.3 Vocabulary of Dynamics Processors

In scientific research, it is widely accepted that if one needs to master a subject quickly, learn the associated vocabulary and industry terms. The same is true with audio recording and production. Most manuals and text books assume a prior level of knowledge. We hope this section will help you to get the most benefit from your independent study in the world of music production.

Compressors

- *Threshold.* The Compressor threshold sets the level at which compression begins. When the signal is above the threshold setting, it becomes 'eligible' for compression. Basically, as you turn the threshold knob counterclockwise, more of the input signal becomes compressed. (If you have a ratio setting of greater than 1:1.)

- *Ratio.* Ratio sets the compression slope. This is defined as the output level versus the input level. For example, if you have the ratio set to 2:1, any signal levels above the threshold setting will be compressed at a compression ratio of 2:1. This simply means that for every 1dB of level increase into the compressor, the output will only increase 1/2dB, thus producing a compression gain reduction of 0.5dB/dB. As you increase the ratio, the compressor gradually becomes a limiter.

  A limiter is defined as a processor that limits the level of a signal to the setting of the threshold. For example, if you have the threshold knob set at 0dB, and the ratio turned fully clockwise, the compressor becomes a limiter at 0dB. This means that the signal will be limited to an output of 0dB regardless of the level of the input signal.

- *Attack*. Attack sets the speed at which the compressor 'acts' on the input signal. A slow attack time (fully clockwise) allows the beginning envelope of a signal (commonly referred to as the initial transient) to pass through the compressor uncompressed, whereas a fast attack time (fully counter-clockwise) immediately subjects the signal to the ratio and threshold settings of the compressor.

- *Release*. Release sets the length of time the compressor takes to return the gain reduction back to zero (no gain reduction) after crossing below the compression threshold. Very short release times can produce a very choppy or 'jittery' sound, especially in low frequency instruments such as a bass guitar. Very long release times can result in an over compressed sound, sometimes referred to as 'squashing' the sound. All ranges of release can be useful at different times however and you should experiment to become familiar with the different sound possibilities.

- *Hard/Soft Knee*. With hard knee compression, the gain reduction applied to the signal occurs as soon as the signal exceeds the level set by the threshold. With soft knee compression, the onset of gain reduction occurs gradually after the signal has exceeded the threshold, producing a more musical response (to some folks).

- *Auto.* Places a compressor in automatic attack and release mode. The attack and release knobs become inoperative and a pre-programmed attack and release curve is used.

- *Gain Makeup.* When compressing a signal, gain reduction usually results in an overall reduction of level. The gain control allows you to restore the loss in level due to compression. (Like readjusting the volume.)

- *Compressor Sidechain*. The sidechain jack interrupts the signal that the compressor is using to determine the amount of gain reduction to apply. When no connector is inserted into this jack, the input signal goes directly to the compressor's control circuitry. When a connector is inserted into this jack, the signal path is broken. This signal can then be processed by an equalizer for example to reduce sibilance (de-essing) in a vocal track. The signal is then returned to the unit via the connector. The signal returned to the sidechain could be that of a narrator or vocalist. In this application, the audio that is passing through the compressor will automatically 'duck' when the narrator speaks or vocalist sings.

Expanders

- *Downward Expansion*. Downward expansion is the most common expansion used in pro audio and recording. This type of expansion applies noise reduction to all signals below a set threshold level.

- *Ratio.* The expansion ratio sets the amount of noise reduction applied to a signal once the signal has dropped below the expansion threshold. For example, a 2:1 expansion ratio attenuates a signal 2dB for every 1dB it drops below the threshold. Ratio's 4:1 and higher act much like a noise gate without the ability to tailor the attack, hold and release times.

Noise Gates

- *Threshold.* The gate threshold sets the level at which the gate opens. Essentially, all signals above the threshold setting are passed through unaffected, whereas signals below the threshold setting are reduced in level by the amount set by the range control. If the threshold is set fully counter-clockwise, the gate is turned off (always open), allowing all signals to pass through unaffected.

- *Attack.* The gate attack time sets the rate at which the gate opens. A fast attack rate is crucial for percussive instruments, whereas signals such as vocals and bass guitar require a slower attack. Too fast of an attack can, on these slow rising signals, cause an artifact in the signal producing an audible 'click'. All gates have the ability to 'click' when opening, however a properly set gate will never click.

- *Hold.* Hold time is used to keep the gate open for a fixed period of time following the signal going below the gate threshold. This can be really useful for effects such as 'gated snare' where the gate remains open after the snare hit for the duration of the hold time then abruptly closes.

- *Release.* The gate release time determines the rate at which the gate closes. Release times should typically be set so that the natural decay of the instrument or vocal being gated is not affected. Shorter release times help to clean up the noise in a signal but may cause 'chattering' in percussive instruments. Longer release times usually eliminate 'chattering' and should be set by listening carefully for the most natural release of the signal.

- *Range.* The gate range is the amount of gain reduction that the gate closes down to. Therefore, if the range is set at 0dB, there will be no change in the signal as it crosses the threshold. If the range is set to -60dB, the signal will be gated (reduced) by 60dB, etc.

2.5 DIGITAL CONNECTIONS AND SYNCHRONIZATION

The FireStudio works with any digital device equipped with ADAT lightpipe I/O or S/DIF I/O. This includes digital mixers and standalone A/D devices like the DigiMax series of microphone preamps by PreSonus.

2.5.1 What is ADAT optical?

"Alesis Digital Audio Tape" or ADAT was pioneered by Alesis to simultaneously record eight tracks of digital audio at one time. The ADAT lightpipe protocol was developed to stream eight channels of 20 or 24 bit audio over an optical cable at 44.1kHz or 48kHz to allow for digital transfers between ADAT digital recorders. Today "ADAT" is used as the standard abbreviation for the ADAT lightpipe protocol and is still used to transfer eight channels of digital audio over a single fiber optic cable. The ADAT digital interface has been adopted by many audio manufacturers as a compact way to transfer digital audio data between devices. Today devices ranging from consoles to synthesizers are equipped with ADAT I/O.

ADAT optical uses the same type of optical cables as the TosLink two channel protocol. These cables can be purchased at your favorite local recording store.

2.5.2 What is S/MUX?

"Sample Multiplexing" or S/MUX is used to transmit high bandwidth digital audio using lower bandwidth technology, i.e. ADAT lightpipe. By using S/MUX, a 96 kHz digital audio stream can be transmitted through a protocol based on lower sample rates. S/MUX works by using demultiplexing to join two or more digital audio channels to represent a single higher bandwidth channel. As discussed earlier, the ADAT lightpipe protocol allows for eight channels of digital audio streaming at 44.1 kHz or 48 kHz. By using S/MUX technology, two channels are joined together to represent one channel at 88.2 kHz or 96 kHz. In this way, you can stream four channels of digital audio at 88.2 kHz or 96 kHz over the same lightpipe connection originally designed to stream eight channels of audio at 44.1 kHz or 48 kHz. The FireStudio is equipped with dual S/MUX technology so that, should you choose, you can record 16 channels of audio at 88.2 kHz or 96 kHz.

2.5.3 BNC Sync and Word Clock

When using multiple devices connected through digital audio formats like S/PDIF, AES/EBU, ADAT or TDIF, it is necessary to synchronize them to a single word clock generator. Word clock is used to keep a perfectly timed and constant bit rate between all synced devices to avoid data errors. A word clock generator creates digital pulses that contain no other data (i.e. audio). These pulses clock the internal oscillators of each device and are essential to avoid frequency drift.

A word clock signal is bundled with the audio data in the ADAT Lightpipe protocol; however, many engineers prefer to keep word clock sync and audio separate from each other. This is where BNC word clock connections come into play and the FireStudio has BNC word clock input and output for just this purpose. You will find BNC cables used to deliver dedicated word clock in many quality studios and broadcast facilities worldwide. BNC cables are rugged, lock into position, and can carry clock signals much farther than the standard optical cable. A BNC word clock cable is a 75 Ω, shielded coaxial cable with standard 'twist-lock' BNC-type connections on each end. *Please note: BNC cables are made in several impedances. The FireStudio requires an impedance of 75 Ω to achieve consistent sync.*

2.5.4 Master/Slave and Multiple Digital Devices

Whether you are using the BNC, S/PDIF or ADAT outputs of the FireStudio to generate word clock or you are using the BNC word clock output of another device as your word clock generator, it is necessary to denote one device as the "master" word clock device to which all other digital devices are synced or "slaved". The FireStudio should perform equally well as a master or a slave in most cases, although syncing it to a lesser quality clock source may affect performance. Not all word clock generators are created equal. The general approach is to determine which device has the best clock from which to reference and to designate that device as the word clock Master. This is done with careful listening and A/B testing.

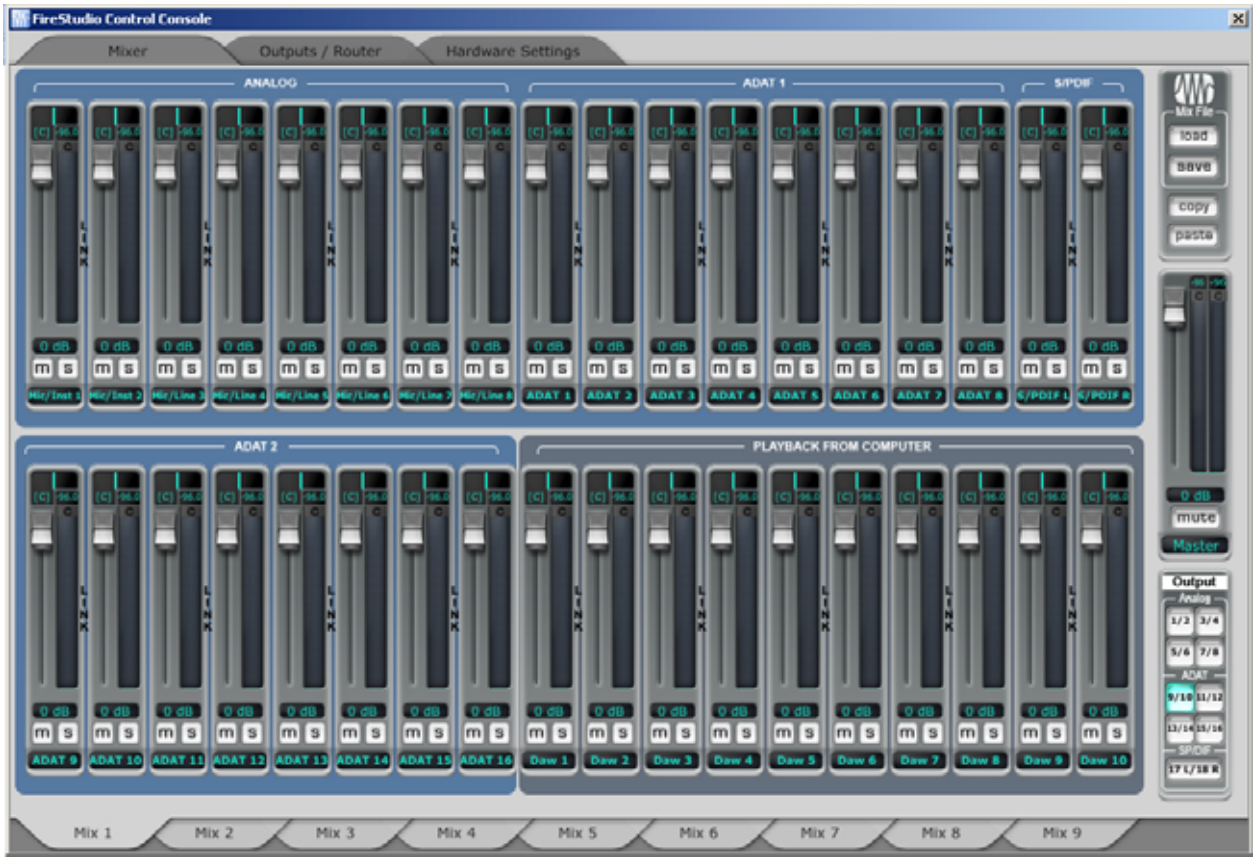
Once you've determined which device is to be your Master clock, you will need to sync the remaining digital devices either through series or parallel distribution or some combination thereof. Of course, if your digital device chain only consists of one master and one slave, syncing the two is as simple as connecting a 75 Ω BNC word clock cable from the output of your master device to the input of the device you are slaving. When working with multiple slaved devices, the job gets a bit more complicated. Series distribution requires that your digital devices have both a BNC word clock input and a BNC word clock output. Parallel distribution uses a BNC "T-connector" attached to the BNC word clock input of each slaved device. This allows the word clock signal to be sent to that device and then sent on to another. A BNC word clock output on the slaved devices is not used or required for parallel word clock distribution. If the last device in the chain does not have a word clock terminate switch, it will require a BNC terminator plug to be attached to the other side of the "T-connector". This helps to stabilize the word clock sync as well as to keep the word clock signal clean. Both word clock terminator plugs and BNC T-connectors can be purchased at most recording supply retailers. For a complete description of parallel word clock distribution as it relates directly to the FireStudio, see section 2.2.2: "Using FireStudio with multiple DigiMax D8s".

A third option for syncing your digital devices is to purchase a high quality dedicated word clock generator. Many engineers believe that using dedicated word clock generators, rather than utilizing series or parallel word clock distribution, enhances the performance of their digital audio devices. A dedicated word clock generator and distribution amplifier exists for one purpose and one purpose only: to be a Master clock. Word clock generators usually have one BNC word clock input and multiple BNC word clock outputs (sometimes TDIF, S/PDIF or ADAT outputs as well to make them compatible with as many types of digital devices as possible). Without a dedicated word clock generator, it is necessary to split the word clock signal generated by the Master device by daisy chaining the slaved devices as described above. Because of this, many engineers feel that the resulting digital audio signals will be of a higher quality when a dedicated word clock generator is used because all the digital devices are receiving the same digital pulse from the same source at exactly the same time.

Whichever approach one uses, it is always advisable to use good quality BNC cables that are not excessively longer than necessary for the job at hand and, as with any audio cabling, it is always good to keep word clock cables separate from AC cable lines or other possible sources of interference.

To synchronize the FireStudio via BNC word clock to an external word clock generator, you will need to run a BNC word clock cable from the BNC word clock output of your external device to the BNC word clock input of your FireStudio. From the FireStudio Control Console's Hardware Settings application, select external BNC sync. In general, however, you will most likely be using your FireStudio as the master clock generator for your external digital devices.

3.1 FIRESTUDIO CONTROL CONSOLE APPLICATION

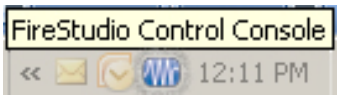


The FireStudio is a revolution in PreSonus interface design because of the flexible routing and mixing capabilities of the FireStudio Control Console (FCC) application. With the FCC, you can:

- Create up to 9 different stereo mixes of inputs and playback streams for flexible zero-latency monitor mixing (i.e. not everyone in the band has to listen to the same thing)
- Create sends and returns for any input to incorporate external processors with zero latency.
- Provide every input with its own hardware output to incorporate a large format console or another recording device
- Send mixes out of analog and digital outputs simultaneously
- Route playback streams to separate physical outputs for live mixing

The FCC consists of three sections: Mixer, Outputs / Router Matrix, and Hardware Settings. We'll go through each of these sections individually.

3.1.1 FireStudio Control Console Application Icon



Once you have successfully installed your FireStudio, the FireStudio Control Console will be available from the Notification Area of your taskbar (typically at the bottom right corner of your screen near the clock).

The FireStudio Control Console icon is red when your FireStudio is disconnected or not installed and blue when the FireStudio is connected and properly installed.

Open the FireStudio Control Console by double-clicking the blue FireStudio Control Console icon or by right-clicking the icon and selecting "Open PreSonus FireStudio Control Console".

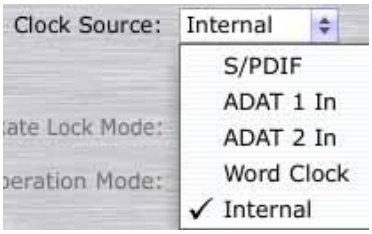
Right-click the FireStudio Control Console icon and select "Quit" to completely exit the FireStudio Control Console application and remove it from your Taskbar. The FireStudio Control Console can also be accessed from the PreSonus FireStudio folder in the Start Menu.

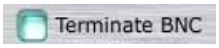
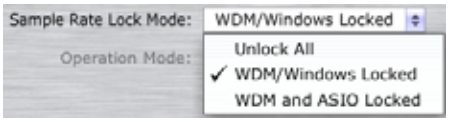
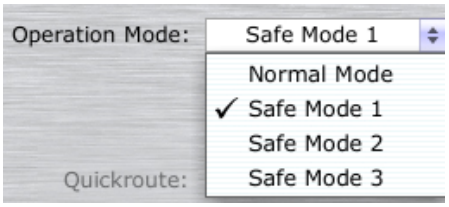
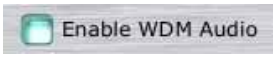
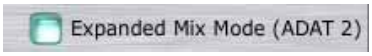
Mac OS X users will find the FireStudio Control Console application in the Applications folder. It's recommended that you move it to your Dock for easy access.

3.2 Hardware Settings tab

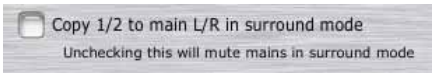
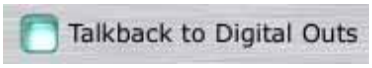
This is where you can set all the basic controls for your FireStudio. We will be including the features that are enabled when an MSR is connected.

3.2.1 FireStudio Settings

▪ **Sample Rate Selector** 	**Changes FireStudio Sample Rate** You can set the sample rate to 44.1, 48, 88.2, or 96kHz. When set at a rate higher than 48kHz, the ADAT/dual SMUX I/O is limited to 8 total channels and requires both ADAT ports. A higher sample rate will increase the fidelity of the recording. It will also increase the subsequent file size and the amount of system resources necessary to process the audio.
▪ **Buffer Size Selector (PC only)** 	**Changes the FireStudio Buffer Size** You can set the buffer size from 64 to 1024 samples. The buffer size (or latency) is the roundtrip time it takes audio data to be converted from analog to digital and back to analog. As a general rule, the higher the buffer size, the better the system performance, but the less playable things like virtual instruments become. In general, 512 samples (11-12 milliseconds) will provide you with a large enough buffer for optimum system performance, but low enough to be unobtrusive. You should set your buffer size and sample rate prior to launching your host application. On Macintosh, the buffer size is set from inside your host application.
▪ **Clock Source Selector** 	**Changes the FireStudio Clock Source** The clock source setting will determine where the FireStudio is receiving word clock information. This keeps the FireStudio in sync with other digital devices. You can choose between Internal, ADAT 1, ADAT 2, SPDIF, or Word Clock. In general, you will want the FireStudio to be your Master Clock so you will set the clock to "Internal". This setting also means that your FireStudio is generating word clock and sending this information out of all of its digital outputs (ADAT outs 1 & 2, SPDIF out, and BNC Word Clock out). If you want the FireStudio to receive sync from an external device, choose the digital source to which the word clock master is connected. The FireStudio's sync light will flash from blue to red. When the FireStudio is in sync, the light will be blue. Please note: *When slaved to an external clock, the FireStudio will not change its sample rate to match automatically. This can cause it not to sync. If your FireStudio is not syncing to an external source, make sure that both your Master device and the FireStudio are set to the same sample rate*.

▪ **Terminate BNC** 	**Terminates BNC:** Check this box to terminate the BNC Word Clock input. This may be necessary to achieve stable sync depending on the external device to which you are syncing. If this device is not self terminating, use this box.
▪ **Sample Rate Lock Mode (PC Only)** 	**Sets which types of applications can change the sample rate** ▪ ***Unlock All.*** Any WDM or ASIO application can change the sample rate ▪ ***WDM/Windows Locked.*** Only ASIO applications can change the sample rate ▪ ***WDM and ASIO Locked.*** Only the FireStudio Control Console can change the sample rate
▪ **Operation Mode** 	**Changes how the FireStudio driver's buffer size is set** ▪ ***Normal Mode.*** Input and Output buffers are both identical to the Buffer Size setting ▪ ***Safe Mode 1-3.*** Increases the ouput buffer size for added stability
▪ **Enable WDM Audio (PC only)** 	**Enables or disables WDM audio in Windows** Disabling WDM Audio is useful when the FireStudio is dedicated for DAW use only.
▪ **Expanded Mix Mode** 	**Allows access to all 26 inputs (44.1 and 48kHz only)** Enabling Expanded Mix Mode replaces DAW playback streams 9-16 with ADAT 2 inputs 17-24. This allows you to create zero latency monitor mixes with all 26 inputs plus 8 individual playback streams from your DAW application.
▪ **Quickroute** 	**Allows you to route a monitor mix to the desired output from the Mixer tab** ▪ ***Digital Mode.*** When Digital Mode is selected you will have the option to route the selected monitor mix to ADAT 9/10, 11/12, 13/14 15/16 or SPDIF 17/18 ▪ ***Monitor Mode.*** When Monitor Mode is selected you will have the option to route the selected Mix to the Main Outputs or any of the headphone outs. You will also have the option to have the mixes automatically route to one of these same outputs as you tab through your 9 monitor mixes *The Quickroute features will be discussed in greater detail in section 3.4.3*
▪ **Reset Names to Defaults** 	**Resets user-defined names** Clicking this button will reset all user-defined names for Mixes and Inputs in the FireStudio Control Console to factory defaults.

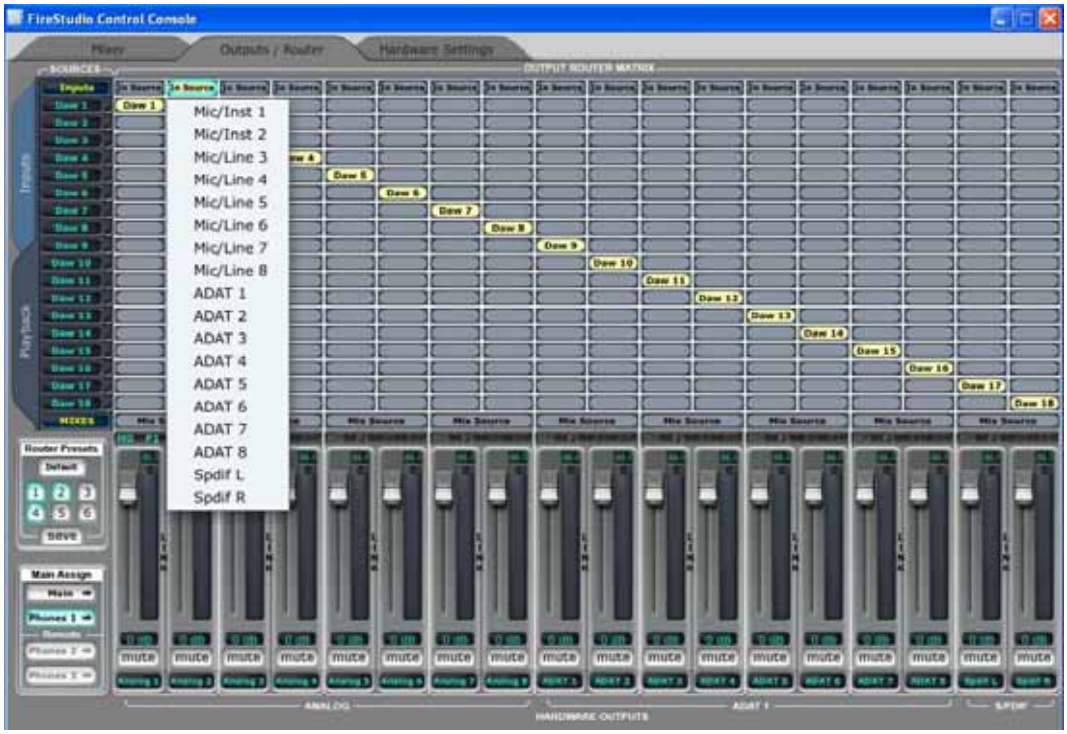
3.2.2 MSR Settings

▪ **Copy 1/2 to Mains in surround mode** 	**Allows you to use mute or enable your Main monitors in surround mode** Enabling this feature allows you to use the monitors connected to Main L/R as your front left/right pair in surround mode without having to recable anything. If you use a different set of monitors for stereo mixing, disabling this feature will automatically mute them when you switch into surround mode
▪ **Quickroute** 	**Sends the talkback mic to all available outputs on the FireStudio** When this feature is enabled, the talkback mic on the MSR will be routed to the SPDIF and ADAT outputs (ADAT 1 output only in 44.1kHz and 48kHz). If you are using your digital outputs for something other than additional monitoring, you should disable this feature.

3.2.3 Version information

▪ **Version Information** 	**Displays information about your FireStudio** This section displays the current Firmware and FPGA version of your FireStudio as well as the version number of the FireStudio Control Console application.

3.3 Output / Router Matrix



True to its name, the Outputs / Router Matrix is the section of the FireStudio Control Console where you have full control of the physical outputs of your FireStudio. The Matrix Router allows you to create custom routing of inputs and playback streams to any output. With the Outputs / Router Matrix you can:

- Route a monitor mix to a stereo output
- Assign every input and playback stream to its own output
- Do quick AD/DA conversion on the fly

The Matrix Router can do it all quickly and easily. This is the virtual patch bay for your FireStudio recording environment.

A quick note on playback streams: *Wherever you see "DAW" in the FireStudio Control Console, this is a playback stream from your host application (or DAW). Traditionally, if you want to route a track in your DAW to a physical output on your interface, you assign this output in your host application. Because the FireStudio Control Console provides much more flexible routing, you can now route this same track to one output or every output, all by itself or as part as a mix.*

For example, if you choose "FireStudio Out 4" for a track or virtual instrument in your DAW, FireStudio Out 4 is not a physical output. It is a virtual output. This track is now "DAW 4" in the FCC. This track can still be routed to the FireStudio's physical out 4, but you can also route to output 2 or 16 and include it in the monitor mix assigned to outputs 7 and 8. But we'll get to that a little later.

Please note: ADAT 2 outputs (FireStudio outputs 17-24) can be directly accessed from your DAW application, no additional routing is available in the FireStudio Control Console.

3.3.1 Inputs and Playback Stream Routing

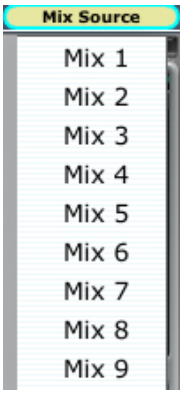
Along the left side of the router matrix, there are two tabs: "Inputs" and "Playback". When the Inputs tab is selected, the matrix will show any inputs that have been routed. When the Playback tab is selected, you can view your playback streams or virtual DAW outputs in the Router Matrix. The Router Matrix patches the selected input or playback stream directly to the specified output.

▪ **Inputs / Source 1 tab** OR	Selecting this tab will allow your to view your Analog, ADAT 1 and SPDIF inputs in the router matrix. If you are in Expanded Mix Mode, this tab will be named "Source 1"
▪ **Input Source** 	When the Inputs tab is selected, you will see a list of your physical inputs (Mic/Inst 1, Mic/Inst 2, etc). Clicking on any box in the row next to each input will assign it to the output at the bottom of that column
▪ **Playback Source Viewer** 	If you assign a playback stream to an output on the Playback or Source 2 tab, this routing will be shown in the row directly above that output.

▪ **Playback / Source 2 tab** OR	Selecting this tab will allow you to view your DAW playback streams in the router matrix. If you are in Expanded Mix Mode, this tab will be named "Source 2". You will be able to view DAW playback streams 1-10 and the ADAT 2 inputs.
▪ **Playback Source** OR	When the Playback tab is selected, you will see a list of your virtual DAW outputs (Daw 1, Daw 2, etc). Clicking on any box in the row next to each playback channel will assign it to the output at the bottom of the column. Remember, these DAW outputs are streaming directly from your host application. If you wish to assign a virtual instrument or a previously recorded track to one of these playback streams, you must first route the output of that track in your host application. (i.e "FireStudio DAW Channels 1-16 and 25-26" in your DAW's output selection translates to "DAW 1-16 and 17-18" in the FireStudio Control Console. Remember FireStudio DAW Channels 17-24 are not accessible through the FireStudio Control Console. Rather, they are directly routed to the ADAT 2 output) If you are in expanded mix mode you will also be able to route your ADAT 2 inputs on this tab.
▪ **Input Source Viewer** 	If you assign an input to an output on the Input or Source 1 tab, this routing will be shown in the row directly above that output.

3.3.2 Mix Routing to Outputs

In addition to being able to patch any input to any output, you can also assign up to 9 different stereo mixes to any pair of outputs as well as assign your main outputs and headphone outputs to mimic a physical output.

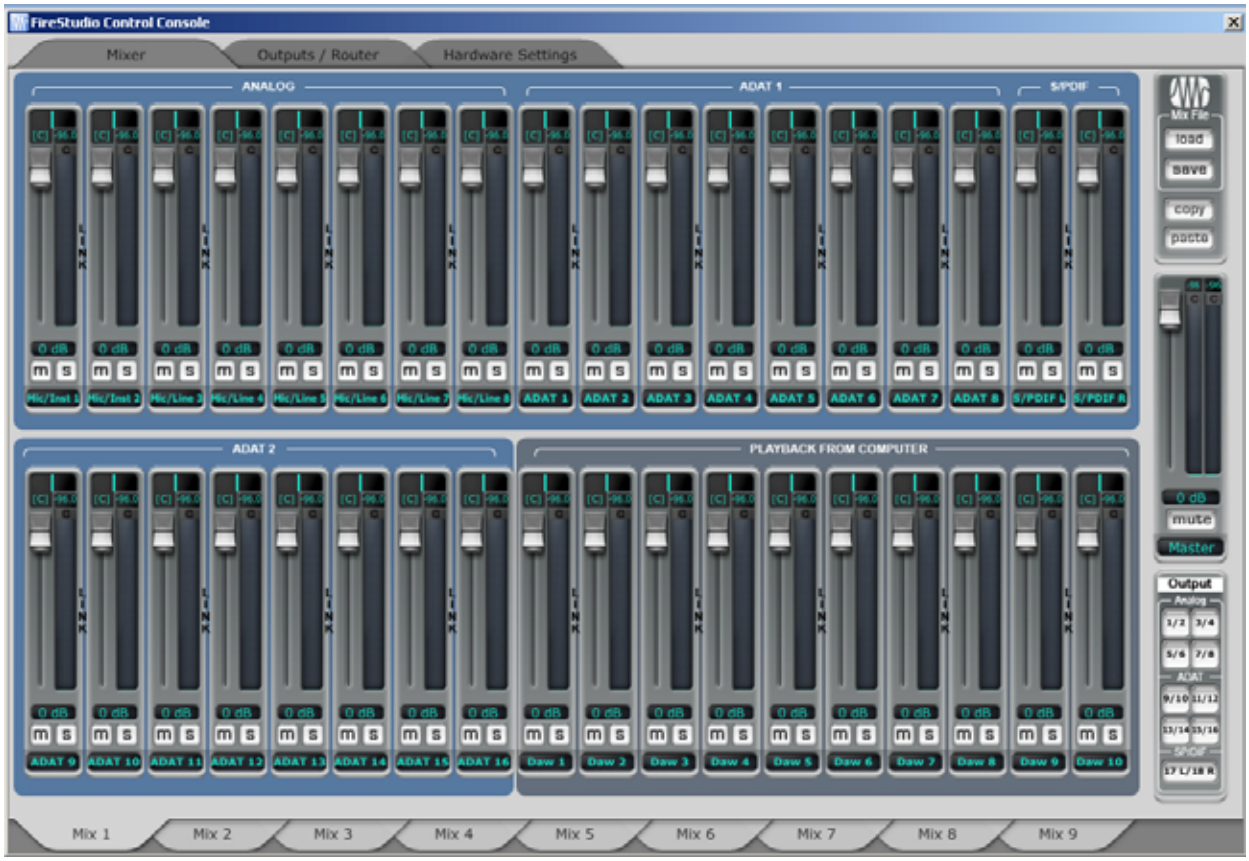
▪ **Mix Source button** 	Clicking on the Mix Source button above the desired pair of outputs will allow you to assign any one of the 9 stereo monitor mixes the FireStudio Control Console allows you to create on the Mixer tab. The Mix Source buttons are visible regardless of which router tab is currently active. When you have assigned a monitor mix to a pair of outputs, the button will show the name of that mix.
▪ **Router Presets** 	The FireStudio Control Console allows you to save six router presets. To save a routing scheme, simply click "Save" and then one of the preset numbers. To recall a preset, simply click on the number. The Default button resets the Router matrix so that each playback stream is routed to its own output.
▪ **Main Assign section** 	The Main L/R outputs as well as each of the headphone outputs can be assigned to mimic any physical output. To assign one of these monitor outputs, click on its button. It will turn blue. Mouse over to the output you wish to mimic, you will see a volume potetiometer appear over the output. Left click this image to audition the output. Double left-click it to assign. Click on the monitor output button again to lock your selection. *Please note: The MSR must be connected for Phones 2 and 3 to be accessible.*
▪ **Main Assign LEDs** 	Once you have assigned the Main or headphone outputs to a particular output pair, an LED will light up above those faders indicating an assignment has been locked. ▪ **M0:** Main Output ▪ **P1:** Phones 1 ▪ **P2:** Phones 2 ▪ **P3:** Phones 3

3.3.3 Output Faders

At the bottom of the Outputs/Router tab, you will see all of your physical outputs enumerated (Analog 1 through S/PDIF R). Each output is visually represented as a fader on a traditional console.

▪ **Channel fader**	The fader controls the volume of the signal on any given output. The signal is metered realtime directly to the right of the fader
▪ **Realtime decibel readout**	Above the meter on each channel strip, you will see a number. This is the dB level of the signal being routed to that ouput
▪ **Fader attenuation**	Below the meter, you will see the fader attenuation (-96 to 0dB). OdB is unity and is achieved by raising the fader all the way up. -96dB represents the fader at its lowest position.
▪ **Clip indicator**	Each meter is equipped with a clip indicator to alert you should your levels get too hot. A red "C" will illuminate when the track has clipped and will remain illuminated until it has been reset. To reset the meter, simply click on the "C".
▪ **Stereo link**	The Link button joins outputs into stereo pairs. To enable this, simply click on the word "LINK". The letters will turn blue. To separate the channels, click the word again. When channels are linked, the volume of both will be controlled with either fader.
▪ **Mute**	The Mute button cuts all signal coming from the specified output. If the channels are linked, muting either of the channels will mute both.

3.4 Mixer tab



The FireStudio Control Console allows you to create 9 stereo monitor mixes. The mixes can be renamed and saved. You can also rename you inputs and playback streams.

It is important to mention that these mixes have no effect on what is being recorded in your host application. This has obvious advantages. However, it is vital to remember that lowering the fader in the FireStudio Control Console on a channel strip that is clipping will not lower the signal that may also be clipping in your host application. You must set the level for the recording using the physical trim pots on the face of the FireStudio. If your signal is too hot in your DAW, it is these pots that will need to be lowered.

The Mixer is made up of 18 inputs and 18 playback streams (Standard Mode) or 26 inputs and 10 playback streams (Expanded Mix Mode), each with its own channel strip. Just like in the Outputs / Router Matrix, you have control both over your physical inputs and you playback streams from your host application.

Across the top row, you will see your Analog, S/PDIF, and ADAT 1 inputs. Across the bottom row, you will see your playback streams. In Expanded Mix Mode, you will also see your ADAT 2 inputs. These playback streams come directly from your host application. Each track or Virtual instrument in your session can be assigned from within your host application to a FireStudio output. This output will be reflected as a corresponding DAW channel in the FireStudio Control Console.

3.4.1 Channel Fader Controls

Each input and playback stream is equipped with its own fader strip. This strip has all the same features as a classic physical console.

▪ **Channel Fader**	The fader controls the volume of the signal on any given output. The signal is metered in realtime directly to the right of the fader.
▪ **Pan Slider**	Above each fader you will see a box with a vertical line. This is your pan control. As you move the pan to the left or right of center, the "C" below it will change to reflect that channel's current pan position.
▪ **Realtime Decibel Readout**	Above the meter on each channel strip, you will see a number. This is the dB level of the signal beign routed to that ouput.
▪ **Fader Attenuation**	Below the meter, you will see the fader attenuation (-96 to 0dB). 0dB is unity and is achieved by raising the fader all the way up. -96dB represents the fader at its lowest position.
▪ **Mute button**	Below each fader is the mute button for that channel. Engaging the mute button will mute that input or playback stream on the current mix only.
▪ **Solo button**	Below each meter is the solo button for that channel. Engaging the solo button will mute all other inputs and playback streams that are not also soloed in the current mix.

▪ **Clip indicator** 	Each meter is equipped with a clip signal. A red "C" will illuminate when the track has clipped. To reset the meter, simply click on the "C". *Please note: The metering in the FireStudio Control Console is completely independent of your host application. If an input signal is clipping while being recorded, lowering the volume in the FireStudio Control Console will not lower the volume in your DAW application. Be sure to monitor your inputs in you DAW as well as the monitor mixes you are creating in the FireStudio Control Console. If a signal is clipping in your DAW, you must adjust the physical trim pot on the face of the FireStudio.*
▪ **Link Button** 	The Link button joins outputs into stereo pairs. To enable this, simply click on the word "LINK". The letters will turn blue. To separate the channels, click the word again. When channels are linked, the volume of both will be controlled with either fader.

3.4.2 Master Fader

Each mix has its own master fader control independent of the Outputs / Router Matrix.

▪ **Master Fader** 	The Master Fader for each Mix has all the same features as the output faders on the Outputs / Router Matrix tab. For a complete detailing of each feature, please review section 3.3.3

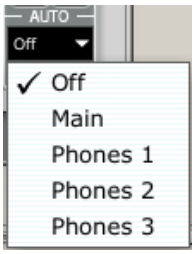
3.4.3 Quick Route section

During a recording session, time is of the essence. The Quick Route section allows you to assign the current monitor mix to the any physical output without bringing up the Outputs / Router matrix. As explained in section 3.2.1, there are two modes for the Quick Route option: Digital and Monitors. When Digital is enabled all 18 of the outputs available in the FireStudio Control Console are visible.

See below:

▪ **Quick Route: Digital** 	Selecting any or all of the output buttons will route the current mix to the enabled output. The buttons will turn blue when enabled.

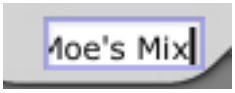
When 'Monitors' is selected on the Hardware Settings tab, the Quick Route section allows you to assign the current mix to any analog output as well as automatically route to either the Main outs or one of the headphone outputs to quickly preview mixes as you scroll through the 9 monitor mixes. Please note: the digital outputs are still accessible and the mixes can still be routed to them on the Outputs / Router Matrix.

▪ **Quick Route: Monitors** 	Selecting any of the output buttons will again route the current mix to the enabled output. ▪ **M0:** Main Output ▪ **P1:** Phones 1 ▪ **P2:** Phones 2 ▪ **P3:** Phones 3 *Please note: The MSR must be connected for Phones 2 and 3 to be accessible.*
▪ **Quick Route: Auto** 	The Auto feature will automatically route whatever mix is currently selected to a chosen monitor output. This allows you to switch between mixes without having to select a new monitor routing each time.

3.4.4 Naming inputs and mixes

The names of inputs, playback streams and mixes on the FireStudio Control Console can be customized. Once these names have been changed on the Mixer tab, the changes will be reflected on the Output / Router Matrix tab as well, making it easy to know at a glance what is connected to each input and for whom each monitor mix was created.

To name your mixes, simply right-click or CTRL-click on the mix name you wish to change, a text box will appear for you to type in the new name:



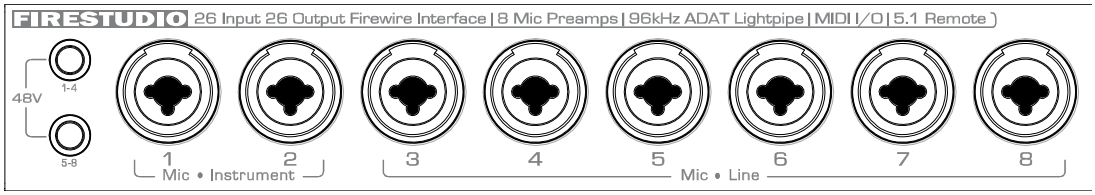
In this way, you can name all 9 monitor mixes:

Once you have named all your mixes, you will see these same custom names on the Output / Router Matrix when you click on a Mix Source button:

You can also name your inputs and playback streams. Each mix will reflect the changes you make to the scribble strip on each channel and, again, these changes will also be viewable on the Output / Router Matrix. To change the inputs, simply double left click on the input name and type the name you wish to use instead.

Renaming your inputs and mixes allows you to fully customize your FireStudio Control Console for faster mixing and routing and lets you get down to the business of making a successful recording.

3.5 FRONT PANEL LAYOUT



- **Microphone Pre-Amplifier.** Your FireStudio is equipped with eight custom designed PreSonus X-MAX microphone preamplifiers for use with all types of microphones including Dynamics, Condensers and Ribbons as well as instruments and line level signals. The award winning PreSonus preamplifier design is a Class A input buffer followed by a dual servo gain stage. This arrangement results in ultra low noise and wide gain control allowing the FireStudio user to boost desirable signal without increasing unwanted background noise.

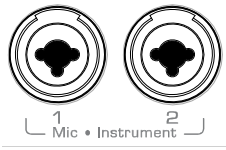
    - ***48 Volt Phantom Power.*** The FireStudio has 48V phantom power available in two groups via push button switches on the front panel. The top button activates phantom power for channels 1-4, the bottom button activates phantom power for channels 5-8.

        - XLR connector wiring for Phantom Power

            - Pin 1 = GND
            - Pin 2 = +48V
            - Pin 3 = +48V

    - ***+22dBu Headroom.*** The FireStudio microphone preamplifier has +22 dBu headroom. This feature gives you wide dynamic range and excellent transient response characteristics.

    - ***Neutrik Combo Connectors.*** Each channel of the FireStudio has a Mic/Line connector using the Neutrik Combo connector. This revolutionary style connector lets you use either ¼" phone or XLR connectors in the same female input. The ¼" input on channels 3-8 are line level inputs. When these inputs are engaged, the microphone preamp circuit is bypassed.

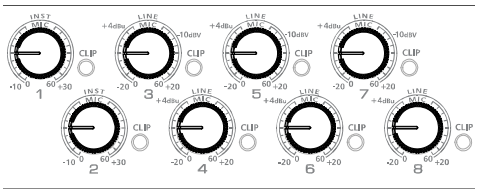


- **Instrument Inputs (Channels 1 and 2).** The ¼" TS connector on channels 1 and 2 are for use with an instrument (guitar, bass, etc.). When an instrument is plugged into the ¼" input, the microphone preamp is bypassed, and the FireStudio becomes an active instrument preamplifier.

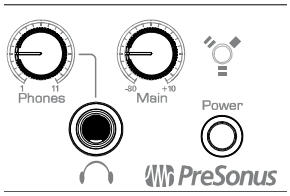
    NOTE: Active instruments are those that have an internal preamp *or a* line level output*. Active instruments should be plugged into a line input rather than into an instrument input. Plugging a line level source into the instrument inputs on the front of the FireStudio not only risks damage to these inputs but also results in a very loud and often distorted audio signal.*

    (In other words, don't plug a line level source into the combo jacks of channel 1 or 2.)

These two channels also feature a send and return on the back panel of the FireStudio. These return inputs can be used as line level inputs for these two channels if instrument preamps are not needed.



- **Input Gain/Trim Control.** These knobs provide the following gain structure:
    - ***XLR Microphone Inputs.*** 54 dB of variable gain (-4 dB to +50 dB)
    - ***TS ¼" Instrument/Hi-Z Inputs.*** 54 dB of variable gain (-4 dB to +50 dB)
    - ***TRS ¼" Line Inputs.*** 32 dB of variable gain (-16 dB to +16 dB trim adjustment)



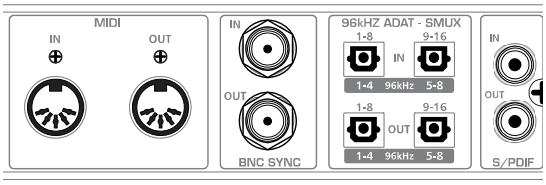
- **Main.** The Main knob controls the output level for the Main Outputs on the back of the FireStudio with a range of -80 db to +10 dB.
- **Phones.** The Phones knob controls the output level of the headphone output on the front of the unit. Notice the volume indicator goes to 11 … use this setting with extreme caution.
- **¼" Headphone Jack.** This is where you connect your headphones to the FireStudio.
- **Red-Blue Power/Sync Light.** This light is a clock source (sync) indicator. It lets you know if your unit is receiving word clock correctly. Word clock is the manner by which digital devices sync frame rates. Proper word clock sync prevents digital devices from having pops, clicks or distortion in the audio signal due to mismatched digital audio transmission.
    - ***Blue.*** FireStudio is correctly synced via FireWire, external ADAT, S/PDIF, or BNC word clock.
    - ***Flashing Red and Blue***. Sync invalid or not present
- **Power button**. Push this button to turn your FireStudio on. The button will illuminate blue. Push it again to power down your FireStudio.

3.6 BACK PANEL LAYOUT

- **Power Adaptor Input.** This is where you plug in the proprietary power supply. The FireStudio features a locking connector to ensure solid power connectivity.

- **FireWire Ports.** There are two standard 6-pin FireWire ports on the back of the FireStudio. Either (and only one) should be used to connect your FireStudio to a vacant FireWire port on your computer. If your computer has a 4-pin connector (commonly found on laptops), you will need a 4 to 6-pin connector to connect your FireStudio to your computer.

  Use the 'extra' FireWire port to connect additional FireWire devices to you computer (such as external Hard Drives, etc).



- **MIDI In and Out Ports.** MIDI stands for "Musical Instrument Digital Interface". However, the MIDI standard goes well beyond just instrumentation and sequencing. The MIDI inputs and outputs allow connection and/or communication with external MIDI equipment. One function of this port is MIDI programming. These can also be used for MMC (MIDI Machine Control) and MTC (MIDI Time Code).

  NOTE: ***MIDI is not audio*** *but is frequently used to trigger or control an audio source (such as a VST plugin or synthesizer). When using MIDI, ensure your MIDI data is correctly sent and received by the appropriate hardware or software instruments. If you are using an external MIDI devices like a synthesizer or a sampler, you will need to connect the audio output of the device to an analog or digital input on the FireStudio in order to record it as audio. Please consult the User's Manual of your MIDI devices for help with MIDI setup and usage.*

- **BNC Input and Output.** These allow the FireStudio to receive and transmit word clock to and from other digital audio devices.

  You will need to select "wordclock" as the Clock Source in the FireStudio Control Console when using the BNC input. A 75ohm BNC word clock cable is required to achieve proper sync.

- **ADAT – S/MUX Input and Output** These are the ADAT – S/MUX connections for your external digital devices. When recording or playing back at 44.1 kHz or 48 kHz, each ADAT I/O will provide 8 of the 16 channels consecutively from left to right. When recording or playing back at 88.2 kHz or 96 kHz, each connection will provide 4 of the 8 channels.

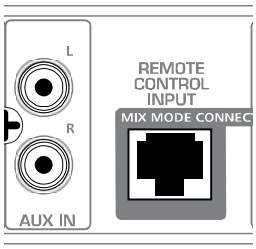
    - ***ADAT 1***: Channels 1-8 (44.1/48 kHz), 1-4 (88.2/96 kHz)
    - ***ADAT 2***: Channels 9-16 (44.1/48 kHz), 5-8 (88.2/96 kHz)

  The ADAT – S/MUX I/O also allows the FireStudio to send and receive Word Clock to external digital devices

  You should select 'ADAT 1' or 'ADAT 2' as the Clock Source in the FireStudio Control Console when using one of these inputs for external sync.

- **S/PDIF In and Out Jacks.** The S/PDIF standard allows two channels of audio to be transmitted at rates up to 24bit/96kHz. The S/PDIF I/O also allows the FireStudio to send and receive Word Clock to external digital devices.

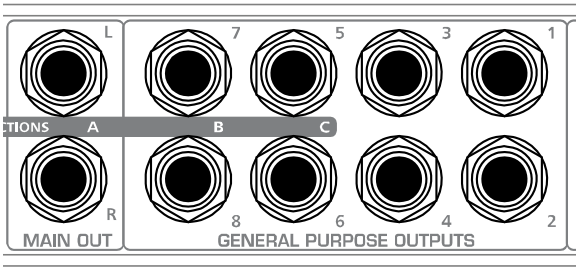
  You should select 'S/PDIF' as the Clock Source in the FireStudio Control Console when using the S/PDIF input for external sync.



- **Aux Input.** This unbalanced stereo RCA input is for use with any device with an analog RCA output.

  This input is only available when the MSR is connected. It allows quick comparison of your internal mix with other playback material (i.e. comparing your mix to a CD, etc). To listen to this input, select the 'AUX' button on your MSR.

- **Remote Control Input.** This is the connection for the optional MSR: Monitor Station Remote. The MSR connects to the FireStudio with a standard CAT-5 (Ethernet) cable.



- **Main Output.** This stereo TRS output derives its source signal from the FireStudio Control Console as described in Section 3.3.2. This output level is attenuated by the Main Level control on the front panel of the FireStudio.

- **General Purpose Line Outputs (TRS Balanced).** These are general purpose line-level outputs. All eight Line Outputs can be accessed in the FireStudio Control Console and can be used for separate mixes, additional speakers, external effects processors, etc.

  Both the Main Output and the General Purpose outputs have additional functionality when an MSR is connected to the FireStudio.

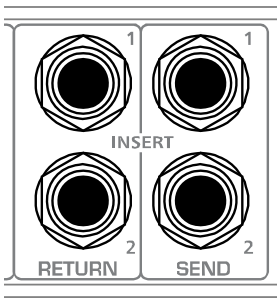
  When using the MSR for monitor switching the outputs are routed as follows:

  - ***Main Output Left/Right.*** Monitor set 'A'
  - ***General Purpose Outs 7/8.*** Monitor set 'B'
  - ***General Purpose Outs 5/6.*** Monitor set 'C'

  When using the MSR for 5.1 surround sound monitoring, the General Purpose outs are assigned as follows:

  - ***General Purpose Out 1.*** Front Left
  - ***General Purpose Out 2.*** Front Right
  - ***General Purpose Out 3.*** Rear Left
  - ***General Purpose Out 4.*** Rear Right
  - ***General Purpose Out 5.*** Front Center
  - ***General Purpose Out 6.*** Sub

  As mentioned in section 3.2.2, the Main Outputs can be assigned to be used separately from the surround monitors, or as the Front Left and Front Right monitors when mixing in surround.



- **Return 1 and 2 (Line Inputs).** These are the line-level inputs for channels 1 & 2. Their gain structure is identical to the ¼" inputs for channels 3-8 and are typically used for signals not requiring preamplification (like CD/MP3 players, effects processors, etc.) These inputs are normalled to the channel 1 & 2 preamp outputs so take precedence over the corresponding Mic/Instrument input on the front of the FireStudio. (i.e. if a cable is plugged into channel one return input on the rear of the FireStudio, the channel one Mic/Instrument input will be inactive until the cable is removed from return input one.)

- **Send 1 and 2 (Preamp 1 and 2 Outputs).** These ¼" jacks are balanced outputs of the preamps on channels 1 & 2. They are half-normalled to the preamp inputs 1 & 2, so plugging a cable into either of the preamp outputs does not interrupt the signal flow from the preamps to the ADC.

  NOTE: The Send outputs are typically used for patching in external signal processors such as compressors, limiters, EQ's, de-essers, etc. to the audio signals on channels 1 or 2. These outputs are not connected to the DAC (digital to analog converters) in any way and so are only capable of routing the input signals from channels 1 & 2. If you need to route recorded audio or input from channels 3-8, you will need to use the FireStudio Control Console to route the input to one or more of the General Purpose Outputs.

4.1 TROUBLESHOOTING

Please note that many technical issues can arise when converting a standard computer into a DAW (Digital Audio Workstation). PreSonus can only provide support for issues that directly relate to the FireStudio interface. It may be necessary to contact the manufacturer of the computer, operating system and/or DAW software to obtain additional technical support. PreSonus does not provide support for issues in regards to operating systems, additional hardware or software. Please check our website, www.PreSonus.com regularly for software information and updates, firmware updates and technical support. Technical assistance may also be received by calling PreSonus at 225-216-7887 Monday through Friday between the hours of 9am and 6pm Central Standard Time. PreSonus Tech Support can also be contacted during this same time via email at techsupport@presonus.com.

Pops and Clicks

Pops and clicks in either your Input or Playback audio are typically caused by momentary losses of synchronization between the FireStudio and its clock source. This can be due to a FireWire card not suited for firewire audio devices or to a lack of CPU power or available memory. Try closing all unnecessary programs, increasing the Buffer Size in the FireStudio Control Console and optimizing your operating system for audio.

Pops and clicks exclusive to the digital inputs or outputs are typically caused when the FireStudio is not digitally synced to the external digital device. If this is the case, ensure your FireStudio and external digital device are properly synced. *Please consult your external device's User's Manual for more details on syncing it to your FireStudio.*

Flashing Sync Light

The FireStudio's Sync light flashes red and blue when it is not synced. Check the firewire cable to ensure it hasn't become disconnected. If you are using an external digital device and your Clock Source is set to the correct digital input, set the sample rate in the FireStudio Control Console to the same sample rate as that of your master clock. If you are not using an external device to generate word clock, open the FireStudio Control Console and set the Clock Source to "Internal".

Audio Drop Outs

Audio drop outs can occur when the speed of your processor cannot buffer audio fast enough. Increase your Buffer Size in the FireStudio Control Console and try the different Operation Modes on the Hardware Settings tab.

No Input Signal on Channel 1 or 2

1) Check your mic cable. This should typically be an XLR connection.
2) Make sure the microphone does not require phantom power. If it does, press the 48V button.
3) Make sure nothing is plugged into the line input on the rear of the FireStudio. The return inputs on channels 1 and 2 take precedence over the combo input on the front of the unit for channels 1 and 2. If a cable is plugged into the return input on channel 1, then the mic/instrument input on channel one will be inactive until the cable is removed from return input 1.

Recommended Firewire Chipsets

The FireStudio will work with a wide range of firewire cards and configurations. However, due to the plethora of firewire chipsets currently on the market, it is not possible to thoroughly test each one for compatibility. Most users will not need to alter their current PC configuration to use their FireStudio. The following firewire chipsets, however, have been tested and approved for use with the FireStudio:

- Agere/LSI FW323-06
- TI TSB43AB23
- VIA VT6308
- VIA VT6306 (on some older motherboards this chip-set will only support a limited number of playback channels: 32 channels at 44.1 or 48kHz; 16 channels at 88.2 or 96kHz)

These chipsets are available on a wide range of motherboards and PCI/PCMCIA firewire cards. Here is a brief list of manufacturers and models of PCI firewire cards:

- Lucent IEEE 1394a
- Syba SD-LUD-4F
- HP PA997A
- Eforcity PCRDFW31C0N1
- Sabrent SBT-VT6306
- Micropac Technologies SBT-VT6306
- SIIG NN-400012-S8

Please note: manufacturers may change chipsets at any time, so it is recommended to verify these models still employ a chipset in the above list prior to purchasing.

At the time of the publication of this manual, PreSonus only recommends the following Express cards:

- ADS Pyro 1394a
- StarTech EC13942

Known Incompatible Hardware

- ***ATI RADEON 9000/9001 IGP video chipset.*** Symptoms are consistent click and pops during audio playback. This video chipset is only found in PC laptops and is entirely integrated as the computer's only video controller. PreSonus strongly recommends that users do not purchase a system with this chipset as there is currently no workaround for this incompatibility.

- ***USB/firewire and s400/s800 combo cards.*** Symptoms usually include no audio recording/playback even though the device will install and sync, as well as erratic audio performance. Rarely combo cards will prevent the device from installing or achieving a stable sync. We recommend firewire cards that have s400 firewire connections only.

- ***Firewire cards with NEC chipsets.*** Symptoms include installation issues, erratic audio and extraneous static and noise.

- ***Motherboards with nForce4 chipsets***. Symptoms include reduced to very poor performance especially if using the onboard firewire connection. Installing a PCIe (not PCI) FW400 only card with an approved chipset is a known workaround, but may not allow full performance.

4.2 SPECIFICATIONS

Microphone Preamp (XLR Balanced)

Frequency Response (+0, -0.5dB) ..... 20Hz to 50kHz
Frequency Response (+0/-3.0dB) ..... 20Hz to 150kHz
Input Impedance ..... 1600 Ohm
THD + N (unwtd, 1kHz @ +4dBu Output, Unity Gain) ..... < 0.003%
EIN (unwtd, 55dB Gain, 150 Ohm Input, 20Hz to 22kHz) ..... -126dBu
S/N Ration (Unity Gain, unwtd, Ref = +4dBu, 20Hz to 22kHz) ..... >101dB
Common Mode Rejection Ratio (1kHz, 55dB Gain) ..... >55dB
Gain Control Range (+ / - 1dB) ..... -6dB to 55dB
Maximum Input level (Unity Gain, 1kHz @ 0.5% THD+N) ..... +17dBu
Phantom Power (+ / - 2VDC) ..... +48VDC

Instrument Input (1/4" TRS, Channels 1 & 2)

Input Impedance ..... 1 Mega Ohm

Line Inputs (1/4" TRS, Channels 3-8)

Frequency Response (+0, -5dB) ..... 20Hz to 50kHz
Frequency Response (+0/-3.0dB) ..... 20Hz to 150kHz
Input Impedance ..... 10 KOhm
THD + N (unwtd, 1kHz @ +4dBu Output, Unity Gain) ..... < 0.003%
S/N Ration (Unity Gain, unwtd, Ref = +4dBu, 20Hz to 22kHz) ..... >101dB
Gain Control Range (+ / - 1dB) ..... -9dB to +12dB
Maximum Input level (Unity Gain, 1kHz @ 0.5% THD+N) ..... +23dBu

Signal Level LEDs

Clip (+ / - 0.5dBu) ..... +18dBu

Digital Audio

ADC Dynamic Range (A-wtd, 48kHz) 107dB
DAC Dynamic Range (A-wtd, 48kHz) 110dB
Bit Depth 24
Reference Level for 0dBFS ..... +18dBu
Internal Sample Frequency Selections (kHz) 44.1, 48, 88.2, 96
BNC Word Clock Output Level (75 Ohm) 4.5V
BNC Word Clock Input Level Range ..... 3.0 to 5.5V

Clock

Jitter ..... <20ps RMS (20 Hz – 20 kHz)
Jitter Attenuation ..... >60 dB (1 ns in => 1 ps out)

Power

Input Voltage Range ..... 18 to 30 VDC
Power Requirements (Continuous) 24W
External Switching Power Supply ..... 90-230VAC / 35W (Factory Configured for Country of Destination)